REQUEST FOR PROPOSALS
FOR
IP-BASED ENTERPRISE COMMUNICATIONS SOLUTION

Town of Fountain Hills
16705 East Avenue of the Fountains
Fountain Hills, Arizona 85268

SOLICITATION INFORMATION AND SELECTION SCHEDULE

Solicitation Title:	**IP-Based Enterprise Communications Solution**
Release Date:	**February 8, 2012**
Advertisement Dates:	**February 15, 2012 – Fountain Hills Times** **February 16, 2012 – Arizona Business Gazette**
NON-MANDATORY Pre-Submittal Conference:	**NOT APPLICABLE FOR THIS SOLICITATION**
Final Date for Inquiries:	**March 7, 2012**
Proposal Due Date and Time:	**March 14, 2012** **4:00 p.m.** (local time, Phoenix, Arizona)
Oral Interviews (if necessary):	**March 28, 2012**
Target Town Council Award Date:	**April 19, 2012**
Anticipated Agreement Start Date:	**April 23, 2012**
RFP Administrator:	**Mike Ciccarone mciccarone@fh.az.gov** **480-816-5171**

*   In the event that a Vendor can not be selected based solely on Proposals submitted, Oral Interviews may be conducted at the Town's sole discretion.

**   The Town of Fountain Hills reserves the right to amend the solicitation schedule as necessary.

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I.   RFP PROCESS; AWARD OF AGREEMENT

1.   Purpose; Scope of Work. The Town of Fountain Hills (the "Town") is issuing this Request For Proposals (this "RFP") seeking proposals ("Proposals") from qualified, licensed firms ("Vendors") to provide and install an IP-based enterprise communications solution for the Town's Civic Center campus, including Town Hall offices and the Community Center, with expansion capabilities for additional sites as may be deemed necessary by the Town (the "System"). The Town desires to select a Vendor that can implement the System, including installation, configuration and testing, by June 13, 2012. In accordance with the Town's Procurement Code, the Town will accept sealed Proposals for the System as more particularly set forth herein.

1.1   Objectives; Background. The Town desires to:

A.   Select a new IP-based communications solution which meets functional, service, cost, management, and supplier reliability criteria.

B.   Select a solution that enables additions, moves and changes at minimal cost.

C.   Select a new voicemail system with auto attendant (embedded or standalone).

D.   Ensure a solid investment with built-in capacity that will handle growth for the next ten years.

E.   Install new communications applications with added functionality, e.g., easy migration path to features such as interactive voice response ("IVR"), speech recognition, computer telephone integration ("CTI"); improved user features for managing communications and mobility; improved messaging; improved contact center capabilities; with minimal disruption to end-user staff.

F.   Select a Vendor that currently supports other customers in the same geographical area.

G.   To select a single Vendor able to provide all System services required.

H.   To select a Vendor that will provide a single point of contact for product information, service and/or support.

1.2   Proposed Configuration. The selected System will provide a VoIP phone system, all hardware necessary, internal voicemail and a UPS battery backup with battery sufficient for a minimum of one (1) hour run time for the Civic Center Complex. In addition, the System shall include:

A.   A teleworker solution with all equipment necessary for installation.

B.   Three (3) enhanced IP phones with DSS/BLF with built-in gig Ethernet switch.

C.   Eighty-five (85) enhanced IP phones with built-in gig Ethernet switch.

D.   Two (2) IP audio conference unit.

E.   Capability to provide dial tone to twenty-five (25) analog devices.

F.   Capacity for a minimum of two hundred (200) voicemail boxes.

G.   N+1 system redundancy.

In order to plan for future expansion of the VoIP system, the Town seeks to select a System that has a documented expansion plan and capacity to add an additional:

A.   Twenty-five (25) enhanced IP phones with built-in gig Ethernet switch.

B.   Four (4) IP audio conference units.

C.   Ten (10) analog devices.

The selected System will utilize existing Town network infrastructure, comprised of Netgear GSM7228PS and GSM7252PS POE 10/100/1000 Ethernet switches interconnected with a 40-Gbps multi-mode fiber backbone between Town Hall and Community Center.

2.   <u>Preparation/Submission of Proposal</u>.   Vendors are invited to participate in the competitive selection process for the Services outlined in this RFP. Responding parties shall review their Proposal submissions to ensure the following requirements are met.

2.1   <u>Irregular or Non-responsive Proposals</u>.   The Town shall consider as "irregular" or "non-responsive" and reject any Proposal not prepared and submitted in accordance with this RFP, or any Proposal lacking sufficient information to enable the Town to make a reasonable determination of compliance to the minimum qualifications. Unauthorized conditions, limitations, or provisions shall be cause for rejection. Proposals may be deemed non-responsive at any time during the evaluation process if, in the sole opinion of the Town:

A.   Vendor does not meet the minimum required skill, experience or requirements to perform or provide the Service.

B.   Vendor has a past record of failing to fully perform or fulfill contractual obligations.

C.   Vendor cannot demonstrate financial stability.

D.   Vendor's Proposal contains false, inaccurate or misleading statements that, in the opinion of the Town Manager or authorized designee, is intended to mislead the Town in its evaluation of the Proposal.

2.2   Submittal Quantities. Interested Vendors must submit **one (1) original** and **four (4) copies (five (5) total submittals)** of the Proposal. In addition, interested parties must submit **one (1) original copy** of the Proposal on a CD-ROM (or electronic media approved by the Town) in printable Adobe or Microsoft Word format (or other format approved by the Town). Failure to adhere to the submittal quantity criteria shall result in the Proposal being considered non-responsive.

2.3   Required Submittal. The Proposal shall be submitted with a cover letter with an **original ink** signature by a person authorized to bind the Vendor. Proposals submitted without a cover letter with an **original ink signature** by a person authorized to bind the Vendor shall be considered non-responsive. The Proposal shall be a maximum of **twenty five (25)** pages to address the Proposal criteria (excluding resumes, Vendor Information Form and appendices requested by the Town, but including the materials necessary to address project understanding, general information, organizational chart, photos, tables, graphs, and diagrams). Each page side (maximum 8 1/2" x 11") with criteria information shall be counted. However, one page may be substituted with an 11" x 17" sheet of paper, folded to 8 1/2" x 11", showing a proposed project schedule or organizational chart and only having information on one side. Cover, back, table of contents and tabs may be used and shall not be included in the page count, unless they include additional project-specific information or Proposal criteria responses. The minimum allowable font for the Proposal is **11 pt, Arial or Times New Roman**. Failure to adhere to the page limit, size and font criteria shall result in the Proposal being considered non-responsive. Telegraphic (facsimile), electronic (e-mail) or mailgram Proposals will not be considered.

2.4.   Vendor Responsibilities. All Vendors shall (A) examine the entire RFP, (B) seek clarification of any item or requirement that may not be clear, (C) check all responses for accuracy before submitting a Proposal and (D) submit the entire Proposal by the Proposal Due Date and Time. Late Proposals will not be considered. A Vendor submitting a late Proposal shall be so notified. Negligence in preparing a Proposal confers no right of withdrawal after the Proposal Due Date and Time.

2.5.   Sealed Submittals. All Proposals shall be sealed and clearly marked with the RFP title, **IP-Based Enterprise Communications Solution**, on the lower left hand corner of the mailing envelope. A return address must also appear on the outside of the sealed Proposal. The Town is not responsible for the pre-opening of, post-opening of, or the failure to open, any Proposals not properly addressed or identified.

2.6.   Pricing. The Vendor shall submit the same number of copies of the Fee Proposal as described in Section I, 2.2 in a separate, sealed envelope enclosed with the Vendor's Proposal. Pricing shall be inclusive of all of the services in the Scope of Work as described in Appendix C, attached hereto and incorporated herein by reference.

2.7.   Address. All Proposals shall be directed to the following address: Town Clerk, 16705 East Avenue of the Fountains, Fountain Hills, Arizona 85268, or hand-delivered to the Town Clerk's office by the Proposal Due Date and Time indicated on the cover page of this RFP.

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2.8. <u>Pricing Errors</u>. If price is a consideration and in case of error in the extension of prices in the Proposal, the unit price shall govern. Periods of time, stated as number of days, shall be calendar days.

2.9. <u>Proposal Irrevocable</u>. In order to allow for an adequate evaluation, the Town requires the Proposal to be valid and irrevocable for **90** days after the Proposal Due Date and Time indicated on the cover of this RFP.

2.10 <u>Amendment/Withdrawal of Proposal</u>. At any time prior to the specified Proposal Due Date and Time, a Vendor (or designated representative) may amend or withdraw its Proposal. Any erasures, interlineations, or other modifications in the Proposal shall be initialed in **original ink** by the authorized person signing the Proposal. Facsimile, electronic (e-mail) or mailgram Proposal amendments or withdrawals will not be considered. No Proposal shall be altered, amended or withdrawn after the specified Proposal Due Date and Time.

3. <u>Cost of Proposal Preparation</u>. The Town does not reimburse the cost of developing, presenting or providing any response to this solicitation. Proposals submitted for consideration should be prepared simply and economically, providing adequate information in a straightforward and concise manner. The Vendor is responsible for all costs incurred in responding to this RFP. All materials and documents submitted in response to this RFP become the property of the Town and will not be returned.

4. <u>Inquiries</u>.

4.1 <u>Written/Verbal Inquiries</u>. Any question related to the RFP shall be directed to the RFP Administrator whose name appears on the cover page of this RFP. Questions shall be submitted in writing or via e-mail by the close of business on the Final Date for Inquiries indicated on the cover page of this RFP or submitted verbally (A) at the Pre-Submittal Conference on the date indicated on the cover page of this RFP (if such Pre-Submittal Conference is held) or (B) after the Pre-Submittal Conference but before the Final Date for Inquiries indicated on the cover page of this RFP. In the event the Town is closed on the Final Date for Inquiries, the Vendor shall submit the question(s) to one of the Town Representatives via e-mail or voicemail. Any inquiries related to this RFP shall refer to the RFP title, page and paragraph. However, the Vendor shall not place the RFP title on the outside of any envelope containing questions, because such an envelope may be identified as a sealed Proposal and may not be opened until after the Proposal Due Date and Time.

4.2 <u>Inquiries Answered</u>. Written questions will be read and answered at the Pre-Submittal Conference on the date indicated on the cover page of this RFP. Verbal or telephone inquiries directed to Town staff **will not be answered**. Within two (2) business days following the Pre-Submittal Conference, answers to all questions received in writing or via e-mail or verbally at the Pre-Submittal Conference will be mailed, sent via facsimile and/or e-mailed to all parties who obtained an RFP package from the Town and who legibly provided their mailing address, facsimile and/or e-mail address to the Town. No questions, submitted in any form, will be answered after the Final Date for Inquiries listed on the cover of this RFP.

5.   <u>Pre-Submittal Conference</u>.   A Pre-Submittal Conference may be held.   If scheduled, the date and time of this conference will be indicated on the cover page of this RFP. This conference may be designated as mandatory or non-mandatory on the cover page of this RFP. Additionally, if the Pre-Submittal Conference is designated as mandatory, failure to attend shall render that Vendor's Proposal non-responsive. Vendors are strongly encouraged to attend those Pre-Submittal Conferences designated as non-mandatory. The purpose of this conference will be to clarify the contents of this RFP in order to prevent any misunderstanding of the Town's requirements. Any doubt as to the requirements of this RFP or any apparent omission or discrepancy should be presented to the Town at this conference. The Town will then determine if any action is necessary and may issue a written amendment or addendum to the RFP. Oral statements or instructions will not constitute an amendment or addendum to this RFP. Any addendum issued as a result of any change in this RFP shall become part of the RFP and must be acknowledged in the Proposal submittal. Failure to indicate receipt of the addendum shall result in the Proposal being rejected as non-responsive.

6.   <u>Payment Requirements; Payment Discounts</u>. Any Proposal that requires payment in less than 30 calendar days shall not be considered. Payment discounts of 30 calendar days or more will be deducted from the Proposal price in determining the low Proposal. The Town shall be entitled to take advantage of any payment discount offered by the Vendor provided payment is made within the discount period.

7.   <u>Federal Excise Tax</u>. The Town is exempt from Federal Excise Tax, including the Federal Transportation Tax. Sales tax, if any, shall be indicated as a separate item.

8.   <u>Public Record</u>. All Proposals shall become the property of the Town. After award of an Agreement, Proposals shall become public records and shall be available for public inspection in accordance with the Town's Procurement Code, except that any portion of a Proposal that was designated as confidential pursuant to Section 9 below shall remain confidential from and after the time of Proposal opening to the extent permitted by Arizona law.

9.   <u>Confidential Information</u>. If a Vendor believes that a Proposal or protest contains information that should be withheld from the public record, a statement advising the Town Representative of this fact shall accompany the submission and the information shall be identified. The information identified by the Vendor as confidential shall not be disclosed until the Town Representative makes a written determination. The Town Representative shall review the statement and information and shall determine in writing whether the information shall be withheld.   If the Town Representative determines to disclose the information, the Town Representative shall inform the Vendor in writing of such determination.

10.   <u>Vendor Licensing and Registration</u>. Prior to the award of the Agreement, the successful Vendor shall be licensed with the Arizona Corporation Commission to do business in Arizona. The Vendor shall provide licensure information with the Proposal. Corporations and partnerships shall be able to provide a Certificate of Good Standing from the Arizona Corporation Commission.

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11.   Certification. By submitting a Proposal, the Vendor certifies:

11.1   No Collusion. The submission of the Proposal did not involve collusion or other anti-competitive practices.

11.2   No Discrimination. It shall not discriminate against any employee or applicant for employment in violation of Federal Executive Order 11246.

11.3   No Gratuity. It has not given, offered to give, nor intends to give at any time hereafter, any economic opportunity, future employment, gift, loan, gratuity, special discount, trip favor or service to a Town employee, officer or agent in connection with the submitted Proposal. It (including the Vendor's employees, representatives, agents, lobbyists, attorneys, and subcontractors) has refrained, under penalty of disqualification, from direct or indirect contact for the purpose of influencing the selection or creating bias in the selection process with any person who may play a part in the selection process, including the Selection Committee, elected officials, the Town Manager, Assistant Town Managers, Department Heads, and other Town staff. All contact must be addressed to the Town's Procurement Agent, except for questions submitted as set forth in Section 4, Inquiries, above. Any attempt to influence the selection process by any means shall void the submitted Proposal and any resulting Agreement.

11.4   Financial Stability. It is financially stable, solvent and has adequate cash reserves to meet all financial obligations including any potential costs resulting from an award of the Agreement.

11.5   No Signature/False or Misleading Statement. Failure to sign the Proposal, or signing it with a false or misleading statement, shall void the submitted Proposal and any resulting Agreement.

12.   Award of Agreement.

12.1   Selection. A Selection Committee composed of representatives from the Town will conduct the selection process according to the schedule listed on the cover page of this RFP. Proposals shall be opened at the time and place designated on the cover page of this RFP. The name of each Vendor and the identity of the RFP for which the Proposal was submitted shall be publicly read and recorded in the presence of witnesses. PRICES SHALL NOT BE READ. The Selection Committee shall award the agreement to the responsible and responsive Vendor whose Proposal is determined, in writing, to be the most advantageous to the Town and best meets the overall needs of the Town taking into consideration the scoring criteria set forth in this RFP. The amount of applicable transaction privilege or use tax of the Town shall not be a factor in determining the most advantageous Proposal. After the Town has entered into an Agreement with the successful Vendor, the successful Proposal and the scoring documentation shall be open for public inspection.

12.2   Line Item Option. Unless the Proposal states otherwise, or unless otherwise provided within this RFP, the Town reserves the right to award by individual line item, by group of line items, or as a total, whichever is deemed most advantageous to the Town.

12.3   Multiple Award. The Town, at its sole discretion, may elect to enter into Agreements with multiple Vendors who are qualified to provide the Services. The final terms and conditions of the proposed Agreement will be negotiated by the Town with the successful offerors.

12.4   Form of Agreement. The selected Vendor will be required to execute an agreement for the project/services in a form acceptable to the Town Attorney. If the Town is unsuccessful in negotiating an Agreement with the highest-scoring Vendor, the Town may then negotiate with the second, then third, highest-scoring Vendor until an Agreement is executed. Town Council approval may be required. The Town reserves the right to terminate the selection process at any time.

12.5   Waiver; Rejection; Reissuance. Notwithstanding any other provision of this RFP, the Town expressly reserves the right to: (A) waive any immaterial defect or informality, (B) reject any or all Proposals or portions thereof and (C) reissue an RFP.

12.6   Protests. Any Vendor may protest this RFP issued by the Town, the proposed award of an Agreement, or the actual award of an Agreement. All protests will be considered in accordance with the Town Procurement Code.

13.   Offer. A Proposal is an offer to contract with the Town based upon the terms, conditions and specifications contained in this RFP and the Vendor's responsive Proposal, unless any of the terms, conditions, or specifications is modified by a written addendum or agreement amendment. Provided, however, that no contractual relationship shall be established until the Vendor has signed, and the Town has approved, an agreement between the Town and the Vendor in the form acceptable to the Town Attorney.

II.   PROPOSAL FORMAT; EVALUATION

Upon receipt of a Proposal, each submittal will be reviewed for compliance with the Proposal requirements by the Selection Committee. Proposals shall be organized and submitted in the format as outlined below. Failure to conform to the designated format, standards and minimum requirements shall result in a determination that the Proposal is non-responsive. Additionally, the Selection Committee will evaluate based upon the criteria as outlined in this document. Points listed below are the maximum number of points possible for each criteria and not the minimum number that the Selection Committee may award. If necessary, the Selection Committee may elect to conduct oral presentations/ demonstrations with selected Vendors and/or request site visits from Vendors still under active consideration, at no cost to the Town. Demonstrations may be requested to be held at a Town facility. The Town is not required to hold such presentations or demonstrations and is not obligated to provide Vendors with such an opportunity.

Section 1: General Information   **10 pts**

A.   One page cover letter as described in Section I, 2.3.

B.   Explain the legal organization of the Vendor. Provide identification information of the Vendor. Include the legal name, address, identification number and legal form of the

Vendor (e.g., partnership, corporation, joint venture, sole proprietorship). If a joint venture, identify the members of the joint venture and provide all of the information required under this section for each member. If the Vendor is a wholly owned subsidiary of another company, identify the parent company. Provide the name, address and telephone number of the person to contact concerning the Proposal.

C.   Identify the location of the Vendor's principal office and the local work office, if different.

D.   Provide a general description of the Vendor that is proposing to provide the Services, including years in business.

E.   Identify any contract or subcontract held by the Vendor or officers of the Vendor that have been terminated within the last five (5) years. Briefly describe the circumstances and the outcome.

F.   Identify any claims arising from a contract which resulted in litigation or arbitration within the last five (5) years. Briefly describe the circumstances and the outcome.

G.   Vendor Information Form (may be attached as separate appendix).

Section 2: Experience and Qualifications of the Vendor   **15 pts**

A.   Provide a detailed description of the Vendor's experience in providing similar services to municipalities or other entities of a similar size to the Town; specifically relating experience with respect to IP-based communication solution implementation.

B.   Provide a list of <u>at least three (3)</u> organizations of a similar size or similar operation to the Town in which work has been performed. This list shall include, at a minimum, the following:

    (i)   Name of company or organization.

    (ii)   Contact name.

    (iii)   Contact address, telephone number and e-mail address.

    (iv)   Type of services provided.

The above information must be current, as this will be used to verify references. Inability of the Town to verify references shall result in the Proposal being considered non-responsive.

Section 3: Key Positions   **15 pts**

A.   Identify at least three full-time key personnel members that will render services to the Town including title and relevant experience; include evidence of equipment certification.

B.   Indicate the roles and responsibilities of each key position. Include senior members of the Vendor only from the perspective of what their role will be in providing services to the Town.

      C.   If a subcontractor will be used for all work of a certain type, include information on this subcontractor. A detailed plan for providing supervision must be included.

      D.   Attach a résumé and evidence of equipment certification, if any, for each key personnel member and/or subcontractor to be involved in this Project. Résumés should be attached together as a single appendix at the end of the Proposal and will not count toward the Proposal page limit.

Section 4: Solution Overview   15 pts

      A.   Briefly describe the Vendor's approach to providing the required System and related services described in the Proposed VoIP System section set forth below, its approach to contract management, including its perspective and experience with respect to installation and implementation of IP-Based Enterprise Communications Solutions , customer service, quality control, scheduling and staff.

      B.   Describe any alternate approaches if it is believed that such an approach would best suit the needs of the Town. Include rationale for alternate approaches, and indicate how the Vendor will ensure that all efforts are coordinated with the Town's general objectives.

      C.   If the Vendor does not manufacture the recommended solution, please provide the manufacturer's name and Vendor/manufacturer relationship. If Vendor does not provide installation, warranty or maintenance services, explain manufacturer/subcontractor responsibilities including future support for proposed solution. Briefly describe the various maintenance programs and options available following the warranty period. Identify which programs and/or options have been proposed. Vendor should include a copy of its standard warranty and maintenance terms as an appendix to its Proposal.

      D.   The Vendor shall include a copy of standard terms and conditions applicable to the proposed System solution as an appendix to its Proposal.

Section 5: Proposed VoIP System   25 pts

      Vendors are requested to complete each item/paragraph in this section by inserting: Comply (Yes/No) or Not Applicable (N/A), and may offer additional description where appropriate. Additionally, Vendors are required to complete the forms provided in Section B of this RFP labeled "Appendix A" and "Appendix B."

      5.1   **IP PBX**

      The System architecture offers converged voice and data using the same underlying technology as the Internet (TCP/IP), with voice being transmitted over local area networks (LAN) and wide area networks (WAN). The solution is comprised of control elements, media gateway elements, applications, and a wide portfolio of desktop IP peripherals for small, medium, and large sites.

SECTION A

The System must use a switched IP core that is highly scalable and delivers reliable, robust switching, routing and call control – fully leveraging IP – while providing access to the PSTN using TDM technology.

For IP-device-to-IP-device communications, e.g., calls between Ethernet IP phones, switching is provided via the Town's IP network. For TDM circuit communications, e.g., a call between an analog phone and the PSTN, switching is handled via a conventional circuit-switched (TDM) bus. The ability to use two different switching technologies means that conversion (or gateway/gatekeeper functionality) is only required between IP and TDM calls.

The System architecture packages the traditional circuit-switched bus, call server, gateway/gatekeeper and real time applications resources within a 19" rack.

Applications include:

- Voicemail.

- A web-based management application tool.

The System is able to support both digital and analog trucks for connection to the PSTN or for connecting multiple sites for systems together.

Networking systems together can be achieved using traditional circuit switched technology or the IP infrastructure.

The System must be able to be used as a standalone application, e.g., IP PBX, VMS, ACD.

5.2   Features

The System delivers a complete set of IP and telephony features. (See Features, Sections 5.11(F) – IP Attendant Console, and 5.12 – Voice Mail System)

5.3   Scalability

The System must support up to 200 users, co-located or geographically dispersed yet still can be managed as a single platform from any location.

The System must provide IP PBX, VMS, ACD, unified communications; speech-enabled application functionality; optional desktop video and video conferencing.

Each control element must be self-sufficient with all processing, memory, DSP, Ethernet access, analog interfaces, and PTSN access resources required for full system operation.

Describe the number of active devices that the proposed System can support simultaneously.

5.4   Station and Network Connectivity

Supported connections must include the options listed below.

A.   Internal Voice Lines

- A 10/100/1000 BaseT Ethernet connection is required for the IP sets to connect through an Ethernet LAN to the System.

- On-premises analog station ports (24V per port) for industry standard DTMF analog telephone sets.

- Off-premises analog lines (48V per port) for industry standard sets or a device where the external loop resistance exceeds 600 ohms or where lightning surge protection is required. The maximum resistance on these must be 1800 ohms or less, and the loop length must be 19,000 feet (5,800 M) on 26-gauge wire.

B.   Analog Trunks

- Loop start

- LS/GS (via peripheral)

- E & M (via peripheral)

- Long line extensions (via peripheral)

- DID and tie trunks

- Centrex

Supported services included such analog devices as phone, trunks, fax machines, and modems. Additionally, resources are required for:

- System fail transfer

- Paging

- Music on hold

C.   Digital Trunking

- T1/D4, E1, R2

- Centrex

D.   IP Networking

5.5   IP Networking Connectivity

IP networking features allow branch-to-branch voice traffic between multiple controllers to be routed over an organization's LAN/WAN infrastructure. The System must support all traditional voice features over IP networking. Benefits include items such as the following:

- Displacement of some voice lines (e.g., Centrex or leased lines).

- A single converged infrastructure.

- Opportunities for converged management of voice and data services.

The System provides the following network related features:

- point-to-point T1;

- dial-up switched over the public network; and/or

- WAN data network providing point to multi-point.

The System features include:

- Loop avoidance.

- Step back on busy/route optimization.

- Portable directory number.

- Busy lamp field.

- Overhead paging.

- Transparent desktop features such as:

  o Transfer

  o Conference

  o Group paging

  o Call forwarding

  o Calling line identification (CLID)

5.6 Remote Connectivity

A. Branch Office. The system must deliver head office features and services for both voice and data at the branch office site including:

- Integration functions.

- Survivability.

- Analog station support.

- Local analog trunking.

B.   Teleworker. The System must deliver head office features for both voice and data at the teleworkers' locations.

5.7   Reliability

The System must provide fully featured call control services using an industry standard real time operating system. In this context, fully featured means all of the features and functionality traditionally associated with a TDM PBX such as multiple levels of call forwarding, message waiting, advisory messages, conference calling, account codes call barring, least cost routing, night service, and so on. **See Section B, Appendix A – Features Compliance Form.**

- The IP switching core must be non-blocking.

- The TDM switching system must be non-blocking.

- Voice traffic must have priority over data and other non-real-time communications traffic.

- The System must support QOS and the standard 802.1p for voice prioritization.

- System-critical MTBF is based on the maximum for a complete system with controllers, peripherals and IP phones. A critical system failure is one that inhibits the system from making calls.

For the following, please list the guarantees:

- System Critical MTBF (redundant): ____ years.

- Mean Time to Repair: ____hours.

- System availability: ____%.

Describe the process and methodology used to calculate the values listed above.

To provide "ten 4s" reliability to the end user, the Town of Fountain Hills understands that following components will need to adhere to 99.99% uptime throughout its LAN.

5.8   Resiliency

The System should provide the ability to configure controllers in a primary controller/secondary controller configuration. The secondary controller must be able to provide all call control services to all phones on the primary controller in the event that the primary controller fails. Phones must be able to auto-register with the secondary controller in the event that the primary controller fails.

The secondary controller must be configurable in a geographically dispersed (i.e., across the WAN) location to provide an even greater degree of protection against disruptions at the primary site.

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A controller must be able to function as the primary controller for a group of phones and act as a secondary controller for another group of phones.

5.9   Powering

All IP devices require power that can be achieved via LAN power over Ethernet or individual power adapters. Outline and confirm availability for the following features or options:

- 802.3af support.

- Additional power sockets.

- Ability to connect un-interruptible power supply (UPS).

- No need to change existing LAN switches.

- Set power delivery options.

- 24V AC power adapter (with AV plug in).

5.10   Product Life Cycle

- Describe the planned product replacement cycle.

- What is the mechanism for hardware upgrades? Describe.

- How will the hardware upgrades be implemented? Describe.

- What is the mechanism for software upgrades? Describe.

- How will the software upgrades be implemented? Describe.

- What is the planned maintenance period (in years) from product introduction? Describe.

5.11   Desktop Sets/Appliances

A.   Technical Standards. Single and multi-line phones will be manufactured in accordance with FCC hearing aid compatibility technical standards contained in Section 68.316, and the Telecommunications Act of 1996.

B.   Analog Phone and Devices. The System must be able to support analog devices and services such as:

- Phones

- Analog trunks

- Fax machines

- Modems

- Paging

- Power fail transfer unit

- Music on hold

C.   IP Phones and Devices. The System must offer:

- 802.1P/Q

- G.711

- G.729

- Single port IP phone

- Message waiting indication

- Hard hold key

- Transfer/conference key

- Volume control

- Line powering or separate power supply supported

- Built-in 10/100/1000 Ethernet switch

- Minimum 5x20 character display

- Up to 10 feature/line keys

- Up to 5 programmable multi-function keys

- Ringing indication

- Dedicated headset port

- Auto selection of prime line

- Directory access

- Redial/call history

D.   Programmable Key Module and Interface. List the number of keys available on key modules that can be attached to the IP sets described above. The keys must be programmable for:

- Feature access

- DSS/BLF appearances

- Group appearances

E. IP Audio Conference Units. The unit must focus on the speaker while virtually eliminating ambient room noise and side bar conversations. The unit must meet requirements for single offices, small meeting rooms, and large boardrooms.

F. IP Attendant Console and Features.

(1) Console Requirements:

- Call handling keyboard with mains power supply

- Application software

- PC compatible (Windows XP or higher)

- Dedicated telephony keypad connected via USB

- Up to 6 primary call handling keys, including:

  o Answer

  o Release

  o Cancel

  o Hold

  o Retrieve

  o Recover last call

- Up to 12 user programmable keys

- Context sensitive volume control – ringer, handset

- Handset and headset sockets

- Optional wrist rest and hand set cradles

(2) Applications must include:

(a) Graphical User Interface. System must offer the following capabilities:

- Calls status bar slowing calls waiting to be answered

- Call source and destination

- Calls on hold

- Console status –day/night service, handset/headset etc.

- Application zone – Phone Book, Directory, Shared Notes, Personal Note Pad.

(b)   Phone Book/Directory. The standard System phone book provides access to all entries in the system telephone directory with the option to search by name, location, or department. When a unique directory entry is identified, a single key press dials the number.

(c)   Shared Notes. Where multiple IP consoles are deployed and connected to a LAN, the shared notes provides a place for sharing information, e.g., day notes, shift hand-over details, absences, and FAQs with appropriate answers.

(d)   Personal Note Pad. The note pad provides a place where the attendant can record messages from callers, reminders, or make notes. Any phone or fax number contained with the text can be selected and dialed from the note pad.

(3)   Attendant Console Features

(a)   Trunk Termination. Allows calls to be held on the console and identifies the trunk group being answered. Calls must queue for the attendant and after a programmable time, overflow to the automated attendant. A call escaping from the automated attendant to the attendant console must receive priority in queue.

(b)   On-line Employee Directory Access. Ability to access the employee directory on line by employee surname, first name, department, or title.

(c)   Attendant Camp-On. An incoming outside call, which the attendant attempts to complete to a busy station, line is "held" on the busy line until it becomes free. When the attendants "camps" this call on a station, the busy station will hear a short tone indicating that a call is waiting.

(d)   Automatic Recall. A call "camped-on" a line will, automatically return to the attendant indicating it is an unanswered "camped-on" call if it is not answered within 20 seconds. Call forwarding invoked at the station line will override this feature.

(e)   Alphanumeric Display. Using either digital or alphabetic designations, the trunk circuit or circuit group to which the attendant is connected will be indicated on the attendant's console.

(f)   Trunk Group Busy Lines. The supervisory console will be equipped with lamps indicating when any trunk group has surpassed an "X" percentage of trunk fill.

(g)   Attendant Call Waiting. A display will indicate to the attendant the number of calls waiting in queue to be answered.

(h)   Automatic Hold. The attendant will be able to move from trunk call to another on the console without using a "hold" button.

(i)   Transfer. The attendant will have the ability to transfer incoming or outgoing calls received at the console.

(j)   Conference. The attendant will be able to conference up to six (6) external and/or internal system stations or trunks in any combination. The conference circuit will be equipped with suitable amplification equipment to minimize audio loss or degradation.

(k)   Serial Call. The attendant will able to manually activate a feature, which will hold or return a trunk circuit, after disconnect by the terminating station.

(l)   Busy Verification or Station. It will be possible for an attendant to verify a station busy.

(m)   Call Park. The attendant will be able to place a call in a "park" state on a designated terminal number until it is retrieved. A maximum of four (4) "park" numbers will be provided, and when placed in a "park" condition, the caller will hear music. If the "park" call is not retrieved with 20 seconds, the call should re-ring the attendant.

(n)   Trunk Testing. The console will be capable of selecting each individual trunk or tie line termination to the DS-0 level for operational testing.

(o)   IVR. Able to interface with an interactive voice response unit (IVR) and deliver full functionality.

- Caller accessing the IVR may press 0 to get the attendant.

- The attendant may send a call to the IVR.

(p)   Speech Recognition. Able to interface with speech recognition technology and deliver full functionality.

(q)   Alarms. The attendant console and the system administrator console will be equipped with lamp indicators denoting the presence of either a major or minor system fault.

(r)   On-Site Paging. Able to interface with an on-site paging amplifier and to activate various zones as required and programmed.

(s)   Music-on-Hold. The IP-PBX must be able to interface with a CD player or satellite audio system to provide Music-on-Hold or Commercials-on-Hold.

5.12   Voice Mail System (VMS)

A.   Basic VMS. The System must include a complete basic VMS. Describe in detail the VMS and its interconnection to the main IP-based voice system. The following forwarded call types must be accommodated by the VMS:

- Internal calls with the IP-PBX

- Analog DID, Centrex, 1FLs, or PRI digital service

- Tie trunk

B.   Standard VMS Features.

(1)   Programming. Must be completed by using the web-based system integration tool, to assign voice mail users and update the directory concurrently.

(2)   Scalability. Confirm support for the following installed and maximum quantities at an access GoS of P.01:

Numbers of Users   _____

Number of Ports   _____

Hours of Storage   _____

(3)   Interface to IP PBX. A VMS is required that interfaces with the IP-based phone system.

(4)   Auto Attendant. The VMS must also provide auto attendant capabilities for extensions on the System including:

- Plays different multi-level greetings during open and closed business hours.

- Provides a company directory that uses extension numbers or names as the dialing method.

- Allows single-digit option extension.

(5)   Multiple Simultaneous Language Support. Must provide multiple simultaneous language support.

(6)   Languages Supported. List languages supported.

(7)   Subscriber Mailboxes:

- Must be password protected.

- Provide tutorial that assists new subscribers with mailbox setup.

- Provide simple message retrieval.

- Provide easy-to-use menus that allow subscribers to send urgent, private, or certified messages.

- Provide notification of waiting messages whether subscribers are in or out of the office.

(8)   Station Dialing. In addition to the menu/route, callers may access an individual station either through the input of the extension number or the input of the called party's last name. A total of 200 names plus 200 extension numbers is required.

(9)   Answer Announcement. Individual personalized greetings of 15 to 30 seconds for each mailbox are required. A user's dictated greeting will only occupy the seconds used, with the remaining time to be pooled.

(10)   Message Taking and System Announcement. A system announcement of up to 60 seconds will be possible. It should be possible for the mailbox owner to input separate greetings for internal calls or external calls received on the system. Using a menu system a caller should be able to select between individual mailboxes using the same extension number.

(11)   DTMF Signaling. The system must be able to receive and generate standard DTMF tone signaling.

(12)   Greeting. VMS calls will be answered on the first ring and be time- and date-stamped.

(13)   Escape. A caller will have VMS escape options before or after leaving a message by dialing "0" to reach an operator or up to four (4) digitals to an extension.

(14)   PSTN Connection Blocking. It will not be possible for a caller connected to the PSTN to be reconnected to the PSTN.

(15)   Security. A caller will not be able passing though the attendant to reach an outside line.

(16)   Distributed Lists. The VMS will contain a minimum of 10 distribution lists of at least 25 names each plus "all broadcast".

(17)   Message Forwarding. Messages may be forwarded to single or multiple destinations with or without introductory comments.

(18)   Message Indicators. Receipt of a message in a mailbox will light a message-waiting lamp or cause a stutter dial tone upon lifting of the handset to indicate a message waiting condition.

(19)   Identification (Pass Code) Code. Users accessing the system will enter six (6) digit pass codes, which must be system validated to provide security.

(20)   Pass Code Change Control. The System should allow user directed pass code changes.

(21)   Message Recovery. Mailbox owners will automatically be given mailbox status upon call-in:

- Number of new messages since last access.

- Number of saved messages existing.

- Upon accessing the messages the user may delete, skip, or save a message.

- Saved VMS messages may only be deleted by the user or the VMS administrator.

(22)   Message Reply. Message reply by the mailbox owner will be possible to another mailbox on the same system by depressing a single key.

(23)   Message Review. It will be possible for a caller leaving a voice mail message to review and edit the message.

(24)   User Controls. The VMS protocol will provide the following user controls:

- Playback messages

- Skip to next message

- Forward/review within the message

- Cancel review

- Replay last message

- Replay faster or slower

- Pause

- Append information

- Forward message (to mailbox or list)

- Create new answer announcement

- Increase playback volume

(25)   VMS Remote Maintenance. The System will be equipped with a remote maintenance port to allow the manufacturer, supplier, or system administrator to connect remotely to perform service or administrative functions.

(26)   VMS Security Features will include:

(a)   Barrier Code. Of at least four (4) digits (alphanumeric) with automatic shut-off should three successive failures occur in trying to gain entry within a two (2) minute timeframe or less. At the time of shutdown, an alarm should appear on the on-site administrative terminal.

(b)   ID Code Input. Successful access of the Barrier Code will require the caller to input an ID Code, preferably four (4) digits.

(27)   Client Functionality. The customer should also have the option of accessing the VMS on the Administrators workstation as a client.

(28)   System Changes. The administrator must be able to add/delete mailboxes, change general recordings and perform other administrative duties while the System is in operation.

C.   Optional Voice Mail Features.

(1)   Redundant CPU. Describe the System's ability to disk-mirror system greetings, and or messages or both.

(2)   Disconnect Detection. The VMS must detect that a caller has hung-up and immediately disconnect and restore the line to service.

(3)   Message Taking and System Announcement. The system announcement of up to 60 seconds will be possible and will be available in the event of a switching system failure.

(4)   Audit Trail. Users may designate a necessary written record of message destination, input time and receipt. This audit trail will be printed on the administrative console along with daily reports.

(5)   Network Capability. Scheduled message transfers between mailboxes on individual VMS at different sites will be possible.

(6)   Pass Code Change Control. The system should allow user directed pass code changes, but it should also have quarterly or system administrator set mandated pass code changes.

(7)   System Operation.   The administrative programs and traffic information trapped should be available during system operation.

(8)   Traffic Reports.   Will be available on customer demand or automatically on a pre-programmed basis of quarter, half or one hour periods or daily and weekly. At a minimum, they will report the following:

- Storage space used for announcement or information mailboxes.

- Message storage space.

- Maximum storage space used during the interval.

(9)   VMS Traffic Reports.

- Will be available on customer demand as stipulated above.

- Report storage space used for announcement.

- Total calls answered.

- Total calls routed.

- Total calls abandoned.

(10)   VMS Redundancy.   The basic voice mail system must provide redundancy for programming and greetings through the IP system management software, which must permit the data to be offloaded onto other storage media.

5.13   System Management, Maintenance Alarms, and Security

A.   IP System Management.   The System should have the following programming tools designed for different levels of users:

- System Administration Tool – provides as web-based interface for trained technicians use to program the System.

- Group Administration Tool – a web-based interface that enables administrators and receptionists to make changes to user information; e.g., hunt groups.

- Desktop Tool – a web-based interface for IP telephone users to program their telephone feature keys.

- Configuration Tool – enables the installer to get a new system up and running. It also facilitates migration of legacy systems to the new IP PBX.

B.   Maintenance Alarms.

(1)   The System should define an alarm as an event that takes place when an anomaly is detected and corrective action is required. All attendants who use the IP PBX consoles should be provided with alarm status information when an alarm is raised. There should be at least three classes of alarms:

- Critical – indicates a loss of service that demands immediate attention. This alarm invokes system fail transfer.

- Major – indicates a fault that affects service to many users. Usually results in major degradation in service and requires attention to minimize user concerns.

- Minor – indicates any fault that does not fall into any of the above two classes. (e.g., single set or single trunk failure).

(2)   An alarm condition is cleared when the fault is resolved.

(3)   The System should have LED circuit indicators on the front of each component that indicates the status of the power, trunk circuits, line circuits, message links, and alarm status (as applicable).

C.   Security. The IP PBX offers comprehensive Toll Control as an integrated part of the Call Control. It allows restriction of user access to trunk routes and/or specific external directory numbers. It also allows Class of Restriction (COR) and Class of Service (COS) features that can substantially reduce the risk of toll fraud. Authorized access to the System tools provides protection for various administration commands from unauthorized users.

D.   IP PBX Management Tool to Control Multiple IP PBX Systems. The System must offer a complete web-based telecommunications management tool that enables customers to:

- Manage multiple IP PBX systems.

- Provide user data administration across multiple systems, including the ability to schedule updates (e.g., integrate network telephone directory with network directory service database, schedule MACs and delete users, audit status of managed devices).

- Provide alarm management with page out capability.

- Provide schedule of maintenance functions such as data upgrades, backup, and restore.

- Provide remote software distribution and install.

- Locate unused directory numbers and unused circuits.

E.   The client needs to be web browser based able to support Microsoft IE.

5.14 Voice Data Network Integration

The IP PBX system should offer LDAP Interface for directory management enabling the IP PBX system to integrate with common directory service applications such as those provided by Microsoft.

5.15 Feature Transparency

The proposed System must successfully integrate existing telephony features described in **Appendix A**.

- List the features from **Appendix A** that <u>will not</u> be available on the proposed System.

5.16 Training

The successful Vendor will provide IP station user training to all users of the IP PBX and printed operating instructions with all IP telephony devices. Instructions will be a combination of visual media and hands-on demonstration done in small groups (e.g., 10 people) at a location acceptable to the Town. Required training is as follows:

- End-user

- Administrative

- Console Operators

- Installation and maintenance

Section 6: Project Schedule: 10 pts

Provide a proposed System implementation schedule showing key project milestones and deliverables. Complete System installation, configuration and implementation must be achieved no later than June 13, 2012.

Section 7: Pricing 10 pts

Vendor shall submit the same number of copies of the Fee Proposal as described in Section I, 2.6 in a separate, sealed envelope enclosed with the Vendor's Proposal with the signature of the representative of the Vendor who is authorized to make such an offer. The Fee Proposal shall list the individual cost for each of the expenses detailed in <u>Appendix C</u> and shall provide in sufficient detail to enable Town staff to evaluate all project and System costs.

Total Possible Points for Proposal: 100

III.   ORAL INTERVIEWS; SCORING

In the event that a Vendor cannot be selected based solely on the Proposals submitted, up to three Vendors may be selected for oral interviews. The selected Vendors will be invited to participate in discussions with the Selection Committee on the date indicated on the cover page of this RFP and awarded points based upon the criteria as outlined below. Vendors may be given additional information for these oral interviews. These discussions will relate less to past experience and qualifications already detailed in the Proposals and relate more to identification of the Vendor's project approach and to an appraisal of the people who would be directly involved in this services for this RFP.

> **Oral Interview**
> | | |
> |---|---|
> | 10 | General Information |
> | 25 | Experience and Qualifications of the Vendor |
> | 20 | Key Positions |
> | <u>45</u> | Proposed Services Provided |
> | **100** | **Total Possible Points for Oral Interview** |

Total Points Possible for this RFP: **200**

IV.   **VENDOR INFORMATION FORM**

By submitting a Proposal, the submitting Vendor certifies that it has reviewed the administrative information and draft of the Professional Services Agreement's terms and conditions and, if awarded the Agreement, agrees to be bound thereto.

VENDOR SUBMITTING PROPOSAL	FEDERAL TAX ID NUMBER
PRINTED NAME AND TITLE	AUTHORIZED SIGNATURE
ADDRESS	TELEPHONE FAX #
CITY STATE ZIP	DATE
WEB SITE: _____	E-MAIL ADDRESS: _____

SMALL, MINORITY, DISADVANTAGED AND WOMEN-OWNED BUSINESS ENTERPRISES (check appropriate item(s):

_____   Small Business Enterprise (SBE)
_____   Minority Business Enterprise (MBE)
_____   Disadvantaged Business Enterprise (DBE)
_____   Women-Owned Business Enterprise (WBE)

Has the Vendor been certified by any jurisdiction in Arizona as a minority or woman-owned business enterprise?

If yes, please provide details and documentation of the certification.

Appendix A: Proposed Features Compliance Form

Feature	Description	Compliance (Yes/No/Not Applicable)
Account Codes — Default	Default Account Codes are entered automatically by the system each time a user dials an external number. They may be used to segregate groups in SMDR for billing.	
Account Codes — Verified and Non-Verified	Verified Account Codes let you access features that are not normally available at a station. These Account Codes can be used at any station to change the CoS and COR. Non-Verified Account Codes let you enter codes on the SMDR record for billing and/or call management.	
Account Codes — System	System Account Codes are automatically out-pulsed by the system when outgoing calls are made on a specialized carrier trunk circuit.	
ACD Extended Agent Groups	This feature lets you program a maximum of 64 agent groups with up to 150 agents in each group. By using the ACD Extended Agent Group feature, you can assign up to 500 agents to each group; however, the maximum number of agent groups is reduced to 32.	
Add Held	Add Held lets you move a call on Hold to another line appearance, form a conference with a call on Hold, or add a call on Hold to an existing conference.	
Advanced ARS	Allows day and time zones, route plans, and ARS assignment to be programmed.	
Advice of Charge	Advice of Charge (AOC) allows the caller to determine the cost of a toll call.	
Analog Station support	The system supports analog stations for both incoming and outgoing calls and message waiting.	
Analog Station Feature Support	The analog station supports switch-hook flash feature codes to support call features, like Call Forward, Do Not Disturb, Transfer, Conf, Hold.	
ANI /DNIS	Automatic Number Identification and Dialed Number Identification Service identify numbers that are transmitted on an incoming trunk.	
ANI/DNIS/ISDN Number Delivery	Automatic Number Identification and Dialed Number Identification Service identify numbers that are transmitted on an incoming trunk.	
Automatic Attendant	Allows an external system environment caller to dial through to an extension without having to go through an attendant.	
Automatic Call Distribution	Consists of four main components: call distribution, agent mobility, management and reporting, feature configuration and administration.	

SECTION B

RADs and Call Distribution	Permits the use of Recorded Announcement Devices (RADs) and a uniform call distribution to hunt groups.	
Attendant Busy-Out (Console)	Attendant Busy-Out (Console) places your attendant console in a busy-out condition (absent status) under certain circumstances. In the busy-out condition, incoming calls are automatically rerouted.	
Attendant Busy-Out (Station)	Attendant Busy-Out (Station) lets you busy-out a specific station by using the attendant console. When you busy-out the station, it cannot be used or accessed.	
Attendant Call Information Display	The Attendant Call Information Display provides the attendant with information about called and calling parties.	
Attendant Call Selection	Attendant Call Selection lets you choose which group of incoming calls to answer first; each group is selected by pressing a soft-key on the attendant console.	
Attendant Conference	Attendant Conference lets the attendant set up one or more conference connections between central office trunks and internal stations.	
Attendant Consoles (Multiple)	Multiple Attendant Consoles can be supported.	
Attendant Console short cut keys	Attendant Console Firmkeys on the console can be programmed as one of the following feature keys: Phonebook, Trunk Status, Alarm, SMDA, Select Option, or blank (no application).	
Attendant Console Status Display	Attendant Console Status Display on each attendant console displays various parameters such as Day/Night Service, Attendant Status, and Alarm Status.	
Attendant Directory Number	Attendant Directory Number lets you dial an attendant directory number (typically "0") to reach the attendant. Separate directory numbers can be programmed for each attendant console.	
Attendant Hold	Hold lets you temporarily suspend a telephone call. While the call is on Hold, you can use the other telephone features.	
Attendant Identity Information Display	Attendant Identity Information Display lets you view the console's prime directory number, the software version, and the console's hold slot number.	
Attendant Language Selection	Enables attendant to choose the language of operation for the attendant console (English or Spanish).	
Attendant Lockout	Attendant Lockout prevents the attendant from re-entering a call once the attendant has released.	
Attendant Messaging	Attendant Messaging lets you activate a message waiting condition on a station from the attendant console. The condition can be queried or canceled by the attendant or by a station user with the appropriate Class of Service.	
Attendant New Call Tone	Attendant New Call Tone notifies you of new calls to the attendant console through an audible indication.	

Attendant Recall	Attendant Recall automatically alerts the attendant when a trunk call has been extended to an idle station and not answered within a specified time-out period or when a call on Hold at the console has not been answered within a selected time.	
Attendant Ringer Control	Attendant Ringer Control lets you mute the attendant console ringer. When the attendant console ringer is muted, incoming calls continue to be indicated by the Call Waiting prompt at the top of the display.	
Attendant Serial Call	Attendant Serial Call automatically returns a call to the attendant console when the caller finishes with the called party.	
Attendant Setup and Cancellation of Station Features	The attendant can setup and cancel certain station features such as Call Forward, Do Not Disturb, Callback, and Reminder.	
Attendant System Login	The attendant has access to system programming functions from the attendant console. To access these programming functions, the attendant must log on.	
Attendant Tone Signaling	Attendant Tone Signaling lets the attendant send tones over the circuit once a call has been established.	
Attendant Trunk Group Busy Status	Attendant Trunk Busy Status Display lets you display and/or print the busy status of the system trunk groups from the attendant console.	
Auto-Answer	Auto-Answer lets you automatically answer calls that ring your Prime line.	
Auto-Hold	Auto-Hold lets you automatically place an active call on Hold when you press a line key to originate or receive another call.	
Automatic Route Selection(ARS)	Automatic Route Selection (ARS) simplifies local and long distance dialing by automatically selecting the most convenient and cost-effective route and by inserting and/or deleting the proper routing digits.	
Broker's Call	Broker's Call lets you temporarily suspend a telephone call while you originate a new call. Once the new call has been established, you can alternate between the two calls.	
Busy Dial Through	Busy Dial Through lets you dial a feature access code sequence when a busy condition is encountered. See Callback and Camp-on.	
Call Announce	Call Announce lets you make an announcement to a busy party. The busy party automatically answers the call Hands-free.	
Callback	Callback lets you request that the system notify you when a busy line becomes idle or when an unanswered station goes off-hook and on-hook.	
Callback – System Programmable	Callback – System Programmable lets you program the destination of a matured callback set against a key line or multi-call line group.	

Call Coverage	Call Coverage is provided through a combination of features: Call Rerouting, Call Forward, Do Not Disturb, and Call Distribution.	
Call Duration Display	Call Duration Display provides you with a display of the call duration for incoming and outgoing calls in one minute increments (starting at 0:00) from the beginning of the call to the end of the call.	
Call Forward	Call Forward lets you redirect incoming calls to an alternate number.	
Call Forward - Cancel All	Call Forward - Cancel All lets you cancel all types of Call Forward.	
Call Forward - Follow Me - End Chaining	Call Forward - Follow Me - End Chaining ensures that calls are not further redirected.	
Call Forward - Follow Me - Reroute When Busy	Call Forward - Follow Me - Reroute When Busy forwards the call to the original set's First Alternative Rerouting if the call forward destination is busy.	
Call Forward – Forced	Call Forward - Forced lets you manually redirect an incoming call on your Prime or private line to another number. Options are available for Always, No Answer, On Busy.	
Call Forward – Override	Call Forward - Override lets you bypass any Call Forward condition that is set at the station that you are calling.	
Call Park	Call Park lets the attendant Hold a call so that a telephone user can remotely retrieve the call.	
Call Pickup	Call Pickup lets you answer an incoming call that is ringing at another station.	
Call Privacy	Call Privacy protects a call from audible Call Waiting tones, as the result of a camp-on, and prevents intrusion of any kind (for example busy override).	
Call Rerouting	Call Rerouting lets the system redirect calls to alternate answering points or devices under specified conditions. Call rerouting may be used to redirect calls always (in Day, Night 1, and/or Night 2 mode) or under busy, no answer, or Do Not Disturb conditions.	
Call Waiting Swap	Call Waiting Swap lets you use the switch hook to alternate between two calls when a party is in Call Waiting for your station or when you have a call on Consultation Hold.	
Camp-on (Call Waiting)	Camp-on, or Call Waiting, lets you notify a busy party that you are waiting. An attendant may also put a call through to a busy station to indicate they are waiting. Upon hearing the Call Waiting tone, the busy party can either respond or finish the current call.	
Camp-on Tone Security	Camp-on Tone Security prevents you from hearing Camp-on tone. If any party in a call has this option enabled, no Camp-on tone is returned to anyone in the call.	

Class of Restriction	Class of Restriction (COR) limits a station's access to specified numbers. A station may have three CORs (Day/Night), and the COR may also be changed by using a Verified Account Code.	
Class of Service	Class of Service (COS) defines a station or trunk's feature and timer options. A station or trunk may have three COSs (Day/Night), and the COS may also be changed by using a Verified Account Code.	
Clear All Features	Clear All Features lets you cancel most of the features activated on your extension or another user's extension.	
Conference	Conference lets you connect 3 to 6 people into a single telephone conversation. While you are in a Conference, you can use any of the features that would normally be available during a two-party call.	
Conference Split	Conference Split lets you separate a conference so that you can speak privately with one of the parties. While you are speaking privately with one party, the other party is on Consultation Hold.	
CTI Advanced Telephony	Allows monitoring of the activity and state transitions of extensions.	
CTI Basic Telephony	Permits a host computer application to initiate and clear calls on behalf of an extension on the system through X.409, X.410, and X.25 protocols.	
Dial Tone	You will normally hear continuous dial tone when you lift the handset. You will hear discriminating dial tone (also called interrupted dial tone) or transfer dial tone under certain conditions.	
Dial Tone - Outgoing Calls	The system can provide a pseudo-CO dial tone to prevent possible confusion to station users.	
Dialed Number Editing	Dialed Number Editing lets you edit numbers during dialing.	
Dialing - Conflicting Numbers	The system can differentiate between conflicting numbers such as 1-0-0-0-0 and 1-0-0-0. In this example, if the 5th digit is not dialed within a time-out period, the system assumes that the dialed sequence is complete and makes the call.	
Direct-ln Lines (DIL)	Direct-In Lines (DIL) allow incoming trunks to be assigned to a specific station or hunt group so that calls from the trunk ring the station or hunt group directly.	
Direct Inward Dialing (DID)	Direct Inward Dialing (DID) allows incoming calls on designated trunks to directly access predefined stations (or other answering points) on the system.	

SECTION B

Direct Inward System Access (DISA)	Direct Inward System Access (DISA) lets external callers access the system by using a special trunk. The system sees the DISA trunk as a station with its own Class of Service and Class of Restriction. Calls that enter the system on DISA trunks have access to a variety of system features. In all cases, the DISA trunk can be assigned account codes to provide a high degree of security or additional options.	
Direct Outward Dialing (DOD)	Direct Outward Dialing (DOD) lets you make external calls without the assistance of the attendant.	
Direct Page	Direct Page allows you to page another telephone over its built-in speaker. See Off-Hook Voice Announce.	
Direct Station Select/Busy Lamp Field (DSS/BLF)	A Busy Lamp Field (BLF) allows the status of a directory number to appear on the line status indicator of a telephone set or Programmable Key Module. The monitored device may be on the same system or another system within the same cluster. The key associated with the busy lamp acts as a Direct Station Selection (DSS) key.	
Display Contrast Control	Display Contrast Control lets you adjust the contrast of the alphanumeric display.	
Do Not Disturb	Do Not Disturb (DND) lets you place your set in an apparent busy condition without affecting the outgoing functionality. If someone calls your set while DND is activated, he or she will hear special busy tone.	
DTMF Keypad Support	DTMF Keypad Support lets analog extensions use all 16 keys on a 4x4 DTMF keypad. The additional row of four keys (ABCD) is used to access features in the system.	
Emergency Services – e911	Allows an Emergency Services number to be dialed, which sends a Customer Emergency Services ID (CESID) from the system to the Public Safety Answering Point (PSAP). The CESID is used as a key in the Automatic Location Information (ALI) database to retrieve a database record indicating the precise location of the caller.	
Feature Keys	Feature Keys let you activate features without dialing feature access codes.	
Flash – Calibrated	Flash – Calibrated provides an alternative method of generating a switchhook flash.	
Flash – Switchhook	Flash – Switchhook lets you place a call on Consultation Hold and return to dial tone so that you can invoke station features.	
Flash – Trunk	Flash - Trunk lets you single or double flash a trunk in order to access Centex features.	
Flexible Answer Point	Flexible Answer Point lets station and console users program a night answer point for their incoming trunk calls.	

SECTION B

Ground Button	A Ground Button (Recall Button) lets you place a call on Consultation Hold and return to dial tone so that you can invoke station features. The Ground Button provides an alternative method of producing a Switch-hook Flash.	
Group Page	Group Page lets you page a group of telephones over their built-in speakers.	
Groups - Key System and Multi-call	Key System Groups and Multi-call Groups let multiple telephones share the same extension number. Incoming calls ring all of the idle stations, and the stations stop ringing when one member answers the call.	
Handset Receiver Volume Control	Handset Receiver Volume Control lets you adjust the volume of the handset receiver.	
Hands-free Operation	Hands-free Operation lets you use your telephone without lifting the handset.	
Headset Operation	Headset Operation lets you use a Headset to make and receive telephone calls.	
Hold	Hold lets you temporarily suspend a telephone call. While the call is on Hold, you can use the other telephone features. The call can be either retrieved at the originating telephone or another telephone.	
Hotline	Hotline limits your access to a designated answer point. The system automatically dials the answer point when you go off-hook. The designated answer point can be another station, an attendant, a trunk, or a hunt group.	
Hunt Groups	Hunt Groups let you dial a pilot number and have the call completed to the first idle station in a group of stations. Any station within a Hunt Group may be accessed directly by dialing the station number.	
Intercept Handling	Intercept Handling lets the system control what happens to a call when the call cannot be completed to the required destination. A call may be routed to a tone or to a directory number. Two alternate destinations may be programmed for each condition.	
Interconnect Restrictions	Each peripheral device is assigned an Interconnect Number that is used to restrict one device from connecting with another. Interconnect Restrictions can be used to restrict access to certain trunks, stations, or equipment (i.e. data communications equipment). The restriction is also a function of the direction of the call.	
Language Change	Language Change lets you change the language of the telephone soft-keys and prompts to any one of the following languages: English or Spanish.	
Line Types and Appearances	Line appearance keys are single or shared lines that appear on the telephone programmable keys. There are three types of lines: Prime, Non-Prime, and No Where Prime.	

Line Appearance Ring Types	Each line appearances can be programmed to ring in a different manner.	
Maintenance	The system provides extensive maintenance coverage. All types of peripheral hardware are periodically tested by the system. Maintenance users may also test individual circuits on demand.	
Message Center	An automated messaging service	
Messaging – Advisory	Messaging - Advisory lets you select a short advisory message to show display set users who call your telephone.	
Messaging – Callback	Callback Messaging lets you leave a callback message on a telephone when the called party is busy or does not answer. When you receive a callback message, you can review the message on the display (if applicable) and/or call the sender back.	
Messaging – Dialed	Dialed Messaging lets you leave a message-waiting indication on a telephone. When you receive a message-waiting indication, you call your message taker to accept the message.	
Network Management	Supports network management functions, transparent system clustering, such a Moves/Adds/Changes, Directory updates, Alarms notification, system maintenance and programming access.	
Music	Music lets you listen to the Music On Hold music source through the speaker of the telephone.	
Music On Hold	Music On Hold provides callers with music while they are waiting for a call to be completed. Music On Hold is provided when a call is on Hold, when a call is transferred to a busy party, or when a call is in Call Waiting for a station. The customer provides the music source.	
Voice System Networking	Support both analog and digital networking to create a virtual voice system from a cluster of independent systems.	
Networked ACD	Networked ACD supports ACD functions over a networked environment. Agent groups at different locations (on different systems – IP or PBX) may service calls on the network independently of where the call first entered the network.	
NI3 Calling Name Delivery	NI3 Calling Name Delivery allows the called party to see the name of the caller on the telephone display screen. Nl3 supports both incoming and outgoing calls for the system T9/E1 interface.	
Night Service	Night Service lets you redirect calls to alternate answer points for individual trunks. The answer point used depends on the selected mode of operation.	

Night Service — Automatic	Automatic Night Service places the system into Night service automatically if all attendant consoles are unable to receive calls or if all attendant consoles are inactive and the time-out period has expired.	
Node ID Recognition	Node ID Recognition lets a system in a network determine if an incoming call applies to it or to another system in the network.	
Non-Busy Station	Non-Busy Station lets you program an extension never to return busy tone. This feature is used for special situations such as emergency stations.	
Non-DID Extension	Non-DID Extension allows the system to support sets that are not directly accessible to DID trunks. These calls are transferred to Non-DID Extensions by an Intercept Handling point (such as an attendant or a station).	
Off-Hook Voice Announce	Off-Hook Voice Announce lets you receive a Direct Page during a handset or headset call. See Direct Page.	
Overlap Out Pulsing	Overlap Out pulsing reduces post-dialing delay when trunk calls are originated. Once a route has been determined by ARS, a trunk is seized and dial pulses or tones are out pulsed to the CO. These out pulses are sent before the user has finished dialing to allow faster call setup on analog trunks.	
Override	Override lets you enter a conversation at a busy station or ring a station with Do Not Disturb activated. Before you enter the conversation, all parties receive a warning tone.	
Override Security	Override Security prevents users from using Override on your station.	
Paging	Paging lets you connect to loudspeaker/paging equipment to access individual paging zones or all paging zones simultaneously. Before you are connected to the paging equipment, you will hear a two-second burst of tone.	
Phonebook	Phonebook lets you locate and telephone a system user based on his or her name, extension number, department, and/or location.	
PRI Card	Describes the options supported by the ISDN PRI interface. These options include Min/Max, Automated Min/Max, NFAS (Non-Facilities Associated Signaling), D-channel Backup and Remote LAN Access.	
Printer Support	The system has complete printer flexibility. Any printer port may be programmed for any application. The system supports both system printers for its own applications (such as SMDR and maintenance) and dedicated data communications printers.	

Priority Queuing	Priority Queuing ensures that calls are handled in order of priority. When internal or external callers must wait for calls to be completed, they are placed into a queue and assigned an access priority.	
Privacy Release	Privacy between users who share line appearances in key systems groups is automatic. The privacy release feature allows users to release privacy during a call to allow another member of the key system group to intrude on the call.	
Q.SIG	A protocol that allows you to connect a minimum of two PBXs together to form a virtual private network. Q.SIG supports both incoming and outgoing calls and feature transparency for centralized attendant, centralized voicemail, calling name/number delivery, hold, transfer, conferencing.	
Recall	Recall lets an incoming caller, who has been transferred to an idle station and not answered within a specified time-out period, call back the last party who handled the call. Similar time-out Recalls occur for parties who were transferred to busy stations or who were placed on Hold.	
Redial	Redial lets you automatically dial the last number that you manually dialed.	
Redial - Saved Number	Redial - Saved Number lets you save a number for future dialing. The number remains saved until a replacement number is saved.	
Release	Release lets you forcibly release from an attempted connection to an external party without going on-hook. Release is useful when you encounter a busy or unavailable external party that you are attempting to add to a Conference.	
Reminder	Reminder lets you program your set to ring and provide a message at a specified time within the 24-hour period.	
Ringer Control	Ringer Control lets you adjust the volume and pitch of the telephone ringer.	
Ringing – Discriminating	Discriminating Ringing lets you distinguish between incoming internal calls, incoming trunk calls, tie line calls, and Callbacks by using different ringing patterns (cadences).	
Ringing – Discriminating (Optional)	Optional Discriminating Ringing lets you change the Discriminating Ringing patterns on analog lines so that you hear internal ringing (1 second on and 3 seconds off) for both internal and external calls.	
Ringing Line Select	Ringing Line Select lets you answer any ringing line by going off-hook.	
SMDR - External	Collects data for outgoing and incoming trunk calls.	
SMDR - Internal	Collects data for calls made between stations within the system.	

SECTION B

SNMP Agent	Simple Network Management Protocol (SNMP) governs the management and monitoring of network devices and their functions.	
Programmable Telephone Set Key for Speech Recognition Support	Provides quick and easy 1-button access to the voice recognition system.	
Speaker Volume Control	Speaker Volume Control lets you adjust the volume of the telephone speaker.	
Speed Dial Keys	Speed Dial Keys let you store and dial frequently used numbers by using the personal keys on your telephone.	
Speed Dial – Personal	Personal Speed Dial let you store and dial frequently used numbers by using access codes and index numbers.	
Speed Dial – System	System Speed Dial lets you dial stored system numbers.	
Station Message Detailed Accounting (SMDA)	Station Message Detailed Accounting (SMDA) lets the system accumulate meter pulses (up to an assigned buffer size) that can be read, printed, and cleared from a console. You can collect meter pulses by using either a device (device meter unit accumulation) or an account code (account code meter unit accumulation).	
Station-To-Station Dialing	Station-To-Station Dialing lets you dial any other station directly.	
Loop Test	The Telephone set Loop Test lets you verify the operation of the telephone keys and displays and the integrity of the data path to the switch. The tests are performed from the set (normally after initial system installation).	
Swap	Swap lets you temporarily suspend a telephone call while you originate a new call. Once the new call has been established, you can alternate between the two calls.	
System Access Authorization	Administrative access to the system is controlled by passwords. Different passwords are assigned for each of the five levels of access.	
System Fail Transfer	Provides for analog trunk to analog station cut-through relays in the event of outage of the system.	
T1/D4	Provides support for T1 Channel Associated Signaling.	
TAPI Support	Supports TAPI and TSAPI computer telephony interfaces.	
Tandem Trunking	The voice system can transparently interconnect trunk circuits originating from one CO or voice system and terminating on another (tandem trunking) without attendant intervention.	
Telephone Directory – Privacy Option	Any extension number in the system telephone directory can be designated as private. When an extension number is private, the number is not displayed on other users' telephones.	

Tie Trunk Support	Tie trunks terminate on the attendant console, at station sets, in hunt groups, or on night bells. They may also be arranged as dial-in tie trunks or tandem trunks. Like CO trunks, tie trunks are arranged in groups.	
Toll Control	Toll control allows or denies access to specified routes, CO exchanges, and directory numbers.	
Tone Demonstration	Tone Demonstration lets you hear the tones provided on the voice system.	
Tone Detection	The voice system can detect and analyze call progress tones that originate from the central office during the course of a trunk call.	
Tone Plan Flexibility	Call progress and supervisory tones generated within the system are programmed to meet the requirements of the telephone authorities of the country in which the voice system is installed.	
Traffic Reporting	Provides traffic reports based on system usage to allow better system resource management.	
Transfer	Transfer lets you move a call from one telephone to another. Before completing a Transfer, you can consult privately with the third party and swap between private conversations with each of the parties.	
Transmission Tests	Transmission Tests let you perform the following tests on a trunk: milliwatt test, balance test, and 100 test.	
Trunk Access	Trunk Access lets you access a specific trunk directly. No toll control or ARS checking is done when you use Trunk Access. This feature is used when a maintenance phone is required	
Trunk Answer From Any Station (TAFAS)	Trunk Answer From Any Station (TAFAS) lets you answer any call that rings a night bell. Once you answer the call, you can use any of the features that are normally available at the station.	
Trunk Busy-Out	Trunk Busy-Out lets you busy-out a specific trunk. When you perform a Trunk Busy-Out, the trunk is busied out if it is idle; if the trunk is in use, it is busied out as soon as it becomes idle. When you busy-out the trunk, it cannot be accessed.	
Trunk Group Busy Status	Enables attendants to query the status of trunk groups from the attendant console.	
Trunk Group Hunting	Trunk Group Hunting lets you search for trunk groups in either a terminal or circular pattern. In a terminal hunt group, trunks are always selected in a predetermined order. In a circular hunt group, trunks are selected in a distributed manner (the first free trunk after the fast one used becomes the new first choice).	
Trunk Labels	Trunk Labels may be assigned to individual trunks or groups of trunks. When a trunk call appears at an attendant console, the trunk label and trunk number are displayed.	

Trunk Select - Direct	Direct Trunk Select lets you access an outside trunk for the purposes of originating and receiving external calls. Because the trunk is assigned to a line appearance, you can access the trunk to make or answer calls without the need for trunk access codes.	
Trunk Support	The system supports most public network trunk types (both analog and digital) — Analog, T1/D4, E1, PRI, BRI, R2.	
Uniform Numbering Plan	The system supports the use of a network Uniform Numbering Plan that allows you to use the same digits to reach a station from any location in the network.	
Voice Mail	The system has its own integral voice mail system.	

Voice Mail Interfaces	Most voice processing systems work in conjunction with the voice platform. The system provides the following voice
	processor interfaces:
	      Voice Mail - Digital (T1)/E&M Interface
	      Voice Mail — analog Interface
	      Voicemail — Integral	

Voice mail Soft-keys	Provides the telephone user with a quick and convenient method to access Voicemail. Access to the system is provided through context sensitive soft-keys presented on the telephone set.	

Appendix B: Proposed Standards and Protocols Compliance Form

Standard	Comply (Yes/No)	Comments
1. 802.11b		
2. 802.1d		
3. 802.1p		
4. 802.1q		
5. 802.3		
6. 802.3af		
7. CRTP		
8. DCL		
9. DHCP		
10. Differentiated Services		
11. DNS		
12. DPNSS		
13. Ethernet 10/100/1000MB		
14. E1		
15. FAX – Group 3		
16. FAX – Group 4		
17. G.711		
18. G.726		
19. G.729A		
20. H.323 v2		
21. IP – Internet Protocol v4		
22. ISDN – PRI		
23. ISDN – BRI		
24. IP Precedence		
25. MGCP		
26. QSIG		
27. Q.931		
28. Q.932		
29. RTCP		
30. RTP		
31. RTSP		
32. R2		
33. SIP		
34. SNMP MIB II		
35. TAPI		
36. T1/D4		
37. TSAPI		
38. T.37		
39. T.38		
40. TCP/IP		
41. UDP/IP		

Appendix C: Proposed Pricing – Fee Proposal

Including pricing for all included items such as system hardware, software, design, LAN assessment, implementation, user training, system administration training, and full warranties covering all parts and labor for one year.

Outright Purchase   **$**

Site/LAN Assessment

IP PBX Equipment

IP Phone Equipment

Boardroom/Conference Room IP Conference Phone

VMS (Standalone or Embedded)

Call Recording

Unified Communications

Speech-Enabled Directory

Speech-Enabled Auto Attendant

Mobility/Teleworker

Fax Integration

Installation

Training

Discounts

Tax

Any optional equipment or software

Lease Options

Please describe any leasing, or lease/purchase programs, and list all prices for 3-year, 5-year and 7-year $1.00 buy-out options.